

Minitar MVA11A

VoIP ATA (SIP)

User Manual

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FCC Part 68

This equipment complies with Part 68 of the FCC Rules. On the bottom of this equipment is a label that contains the FCC Registration Number and Ringer Equivalence Number (REN) for this equipment. User must provide this information to the telephone company upon request.

The REN is useful to determine the quantity of devices you may connect to the telephone line and still have those entire devices ring when your number is called. In most, but not all areas, the sum of the REN of all devices connected to one line should not exceed five (5.0). To be certain of the number of devices you may connect to your line, as determined by the REN, you should contact your local telephone company to determine the maximum REN for your calling area.

If the modem causes harm to the telephone network, the telephone company may discontinue your service temporarily. If possible, they will notify you in advance. But if advance notice isn't practical, you will be notified as soon as possible. Users will be advised of your right to file a complaint with the FCC.

The telephone company may make changes in its facilities, equipment, operations, or procedures that could affect the proper operation of your equipment. If they do, you will be notified in advance to give you an opportunity to maintain uninterrupted telephone service.

If you experience trouble with this modem, please contact your dealer for repair/warranty information. The telephone company may ask you to disconnect this equipment from the network until the problem have been corrected or you are sure that the equipment is not malfunctioning.

This equipment may not be used on coin service provided by the telephone company. Connection to party lines is subject to state tariffs.

FCC Part 15

The modem generates and uses radio frequency energy. If it is not installed and used properly in strict accordance with the user's manual, it may cause interference with radio and television reception. The modem has been tested and found to comply with the limits for Class B computing devices in accordance with the specifications in Subpart B, Part 15 of the FCC regulations. These specifications are designed to provide reasonable protection against such interference in a residential installation. However, there is no guarantee that interference will not occur in a particular installation. FCC regulations require that shielded interface cables be used with your modem.

If interference does occur, we suggest the following measures be taken to rectify the problem:

- 1) Move the receiving antenna.
- 2) Move the modem away from the radio or TV.
- 3) Plug the modem into a different electrical outlet.
- 4) Discuss the problem with a qualified radio / TV technician.

CAUTION:

Changes or modifications not expressly approved by the party responsible for compliance to the FCC Rules could void the user's authority to operate this equipment.

Cable connections:

All equipment connected to this modem must use shielded cable as the interconnection means.

Notes:

Operation is subject to the following two conditions:

- 1) This device may not cause harmful interference, and
- 2) This device must accept any interference received including interference that may cause undesired operation.

Chapter 1 Introduction

1.1 Overview

The Minitar VoIP ATA (Analog Telephone Adapter) provides an easy way to make low cost phone calls via the Internet using your existing broadband Internet service. The VoIP ATA is optimized to suit the growing demand for VoIP applications, and it provides a single, highly-integrated and cost-effective solution.

The device has a WAN port to connect to the Internet via a router/modem, and a LAN port to allow a PC to be connected to the device and to also access the Internet via the device and the WAN connection.

The device complies with the SIP v1/v2 standard. You will need to have a VoIP Service Provider (VSP) or SIP server account to make phone calls to other SIP devices or to PSTN phones.

The VoIP ATA device WAN port can be set up as PPPoE client, Static IP or DHCP client to connect to the internet via a modem / router.

It provides a Network Address Translation (NAT) function to allow the Internet WAN connection to be easily shared with a PC or other network device connected to the LAN port..

1.2 Features

- **Key features**
 - STUN (Simple Travel of UDP over NAT)
 - Dynamic DNS
 - DMZ & VLAN Function
 - DHCP (dynamic host configuration protocol) server and client
 - NAT (network address translation)
- **VoIP Features**
 - Voice codecs
 - G.711:64k bit/s (PCM)
 - G.723.1: 6.3k / 5.3k bit/s
 - G.726: 16k / 24k / 32k / 40k bit/s (ADPCM)
 - G.729A : 8k bit/s (CS-ACELP)
 - G.729B : adds AD & CNG to G.729
 - Voice Quality
 - VAD(Voice activity detection),
 - CNG (Comfortable noise generator)
 - LEC (Line echo canceller)
 - Packet Loss Compensation
 - Adaptive Jitter Buffer
 - Call Functions
 - Call Hold
 - Call Waiting
 - Call Forward
 - Caller ID
 - 3-Way Conferencing
 - DTMF Function
 - In-Band DTMF
 - Out-of-Band DTMF
 - SIP Info
 - Phone Function
 - Volume Adjustment
 - Speed dial key
 - Phone book
- **Security**
 - MD5 for SIP authentication (RFC2069/ RFC 2617)
 - Password protected system management
- **Ethernet Interface**
 - Compliant with IEEE 802.3 and 802.3u 10/100 Mbps
- **HTTP Web-Based Management**
 - Firmware upgrade by UI
 - WAN and LAN side connection statistics
 - Configuration of static routes and routing table
 - Password protected access

1.3 PC System Requirements

- 1) Ethernet Network Interface Controller (NIC) RJ45 Port
- 2) Web Browser eg MS Internet Explorer, Firefox, Opera

Chapter 2 Installation

This chapter offers information about installing the VoIP ATA. If you are not familiar with the hardware or software parameters presented here, please consult your service provider for the values needed.

2.1 Checklist

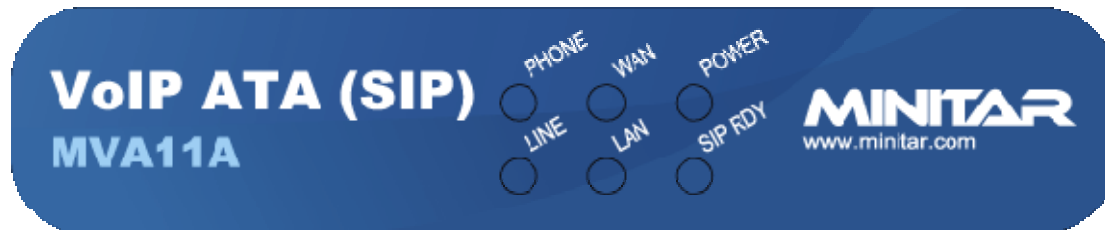
Check the shipping box carefully to ensure that the contents include the items you ordered. If any of the items are missing or damaged, contact your local distributor. The contents of your carton may vary depending on your service provider.

Contents description

- 1) VoIP ATA for home/office use
- 2) Installation and Operation Guide (this publication)
- 3) Power supply with 9V AC / 1 Ampere power adapter
- 4) RJ-11 telephone cable (1.8m)
- 5) Ethernet cable Ethernet category 5 twisted pair cable (1.8m)

2.2 Front Panel LEDs

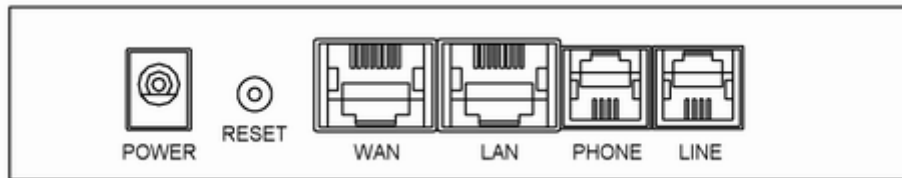
VoIP ATA with 1FXO & 1FXS Ports



| LED | State | Description |
|---------|----------|---|
| POWER | ON | When the router power is on. |
| WAN | ON | When the device is connect to a Router. |
| | Flashing | Data transfer. |
| LAN | ON | When the device is connect to a PC. |
| | Flashing | Data transfer. |
| PHONE | Flashing | When the phone is off hook. |
| | OFF | When the phone is on hook. |
| SIP RDY | ON | The SIP number is registered. |
| LINE | ON | The PSTN Line is connected. |

2.3 Rear Panel Ports

VoIP ATA with 1FXO & 1FXS Ports



| Connector | Description |
|---------------------|---|
| POWER | Power connection to 9VDC supply. |
| RESET Switch | Reset button to restart the ATA when pressed briefly. The ATA will restore default settings when button held pressed until reboot. |
| WAN | The device connection to a router or DSL modem/router. |
| LAN | PC connection to the device. |
| PHONE | Analog phone connection to the device. |
| LINE | PSTN line connection to the device. (Only specific models have this port.) |

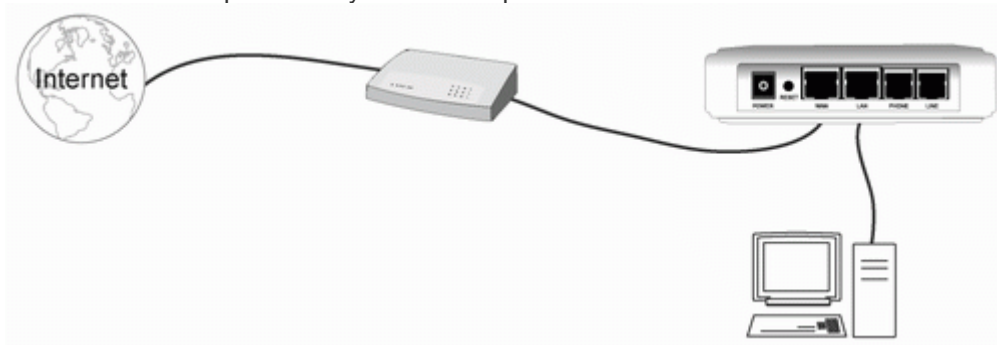
2.4 Hardware Installation

This section describes how to connect and configure the device.

Step 1.

Connect to DSL/Cable Modem or Broadband Router.

Connect the ATA WAN port directly to the LAN port of these devices.



Step 2.

Connect a Phone to the RJ-11 Phone Port.



Step 3.

Connect the PSTN line to the RJ-11 Line port.



Step 4.

Connect the Power Adapter to the Router.

Connect the power adapter to the port labeled POWER on the rear panel of router.



Chapter 3 Configuration

3.1 Determine your connection settings

Before you configure the gateway; you need to know the connection information supplied by your ADSL service provider or just set it as a DHCP client.

3.2 Connecting the Gateway to your network

LAN: Because the ATA Gateway acts as a DHCP server by default, you should set your PC as DHCP Client to automatically accept the IP Address from the Router.

WAN: There are several different operating modes for the WAN connection and you need to know which mode is necessary for your system. These modes are DHCP client, PPPoE, Fixed IP.

3.3 Configuring with Web Browser

The default address of the ATA is **192.168.3.1'**

To configure the router, open your browser, type '**http://192.168.3.1**' into the address bar and click 'Go' to get to the login page. Save this address in your Favorites for future reference.

The default login is **admin** and the password is **admin**

Note: It is advisable to change the administrator password later to safeguard the security of your network.



The screenshot shows a web browser window with the title "Login VoIP". The page has a blue header and a light blue background. The main content area contains the text "Enter your username and password to login" followed by "VoIP server". Below this, there are two input fields: "Username" with the text "admin" and "Password" with masked characters "*****". At the bottom of the form, there are two buttons: "Login" and "Clear".

3.3.1 Setup Wizard – WAN Settings

Setup Wizard 1 - WAN Settings

You can configure the WAN settings on this page.

LAN Mode: ☐ Bridge ☒ NAT

WAN Setting

| | |
|---------------------|---|
| IP Type: | <input type="radio"/> Fixed IP <input checked="" type="radio"/> DHCP Client <input type="radio"/> PPPoE |
| IP Address: | <input type="text" value="0.0.0.0"/> |
| Mask: | <input type="text" value="0.0.0.0"/> |
| Gateway: | <input type="text" value="0.0.0.0"/> |
| DNS Server1: | <input type="text" value="168.95.192.1"/> |
| DNS Server2: | <input type="text" value="168.95.1.1"/> |
| MAC: | <input type="text" value="0005b45000d4"/> |
| Host Name: | <input type="text" value="VOIP_TA1S10"/> |

PPPoE Setting

| | |
|----------------------|--|
| User Name: | <input type="text"/> |
| Password: | <input type="password" value="....."/> |
| Service Name: | <input type="text"/> |

The Setup Wizard is an easy way to set up the VoIP ATA quickly.

In the first step, you can set the LAN mode and WAN IP type. When you are finished changing the settings, click the **Next** button to move to the next page.

3.3.2 Setup Wizard – SNTP Settings

Setup Wizard 2 - SNTP Settings

You can set the SNTP servers on this page.

| | |
|--------------------------|---|
| SNTP: | <input checked="" type="radio"/> On <input type="radio"/> Off |
| Primary Server: | <input type="text" value="time.windows.com"/> |
| Secondary Server: | <input type="text" value="208.184.49.9"/> |
| Time Zone: | GMT + <input type="text" value="08"/> : <input type="text" value="00"/> (hh:mm) |
| Sync. Time: | <input type="text" value="1"/> : <input type="text" value="0"/> : <input type="text" value="0"/> (dd:hh:mm) |

In this page, you can set up the SNTP server for maintaining the time of day clock.

You can setup the primary and secondary SNTP Server IP Addresses to get the date/time information. Also you can set the Time Zone based on your location, and also how often to synchronize the clock.

When you finished the settings, click the **Next** button to move to the next page.

3.3.3 Setup Wizard – Service Domain Settings

In this page, you set up the SIP server settings. You need to input the account details as provided by your VSP or SIP provider. You can register up to three VSP/SIP accounts (Realms).

By default, outgoing calls will be made via the first SIP Realm account.

Notes:

1. Selection of the Realm to be used for outgoing calls may be changed by using the IVR Set Realm function from the phone dial pad (see Sect 3.11) as follows:

Dialling “1*”, “2*”, “3*” sets the default to Realm 1, 2, 3 respectively.
Dialling “0*” sets the default to PSTN

2. Dialling “0*” ahead of a number will switch to the PSTN line (giving second dial tone), and the number will be dialed out on the PSTN line for that call.

Incoming calls may be received via all three Realm accounts if they are correctly registered.

-- To set up each Realm, click on **Active** to enable the Realm dialog box, then you can input the following items:

-- **Display Name:** The name you want to display.

-- **User Name:** The User Name from your ISP.

-- **Register Name:** The Register Name from your ISP (often the same as User Name).

-- **Register Password:** The Register Password from your ISP.

-- **Domain Server:** The Domain Server from your ISP.

-- **Proxy Server:** The Proxy Server from your ISP (often the same as Domain Server).

-- **Outbound Proxy:** The Outbound Proxy get from your ISP. If your ISP does not provide the information, then you can leave this item blank.

When you finished the setting, click the **Submit** button to apply the settings.

You can see the Registration Status in the **Status** item. If the item shows "Registered", then your ATA Gateway has successfully registered to the ISP, you can make a phone call. If not, check the details for correctness.

If you have more than one SIP account, you can follow the steps to register to the other accounts.

Click the **Next** button to move to the next page.

Note Information

This page inform user important information.

Configure OK.

You have to **save** and **reboot** the VoIP to effect those changes.

Save Changes:

After you finish the setup wizard, please ensure that you click the **Save** button to save the settings into the device memory. The device will reboot after saving the settings.

3.4 System Status

System Status

This page shows current status of the system.

| Firmware Version | |
|--------------------------|---------------------------|
| Firmware Version: | Mon Dec 17 17:03:09 2007. |
| Codec Version: | Thu Aug 23 17:43:12 2007. |

| VoIP Service | |
|----------------------|----------------|
| Phone Realm1: | Not Registered |
| Realm2: | Not Registered |
| Realm3: | Not Registered |

| WAN | |
|----------------------|--------------|
| Type: | DHCP Client |
| NAT Type: | Blocked |
| IP Address: | 0.0.0.0 |
| Mask: | 0.0.0.0 |
| Gateway: | 0.0.0.0 |
| DNS Server 1: | 168.95.192.1 |
| DNS Server 2: | 168.95.1.1 |

You can check the System status page for the following information:

- Firmware version
- VoIP Service status
- WAN, and
- LAN status.

3.5 Network Settings

3.5.1 WAN Settings

WAN Settings

You can configure the WAN settings on this page.

LAN Mode: ☐ Bridge ☒ NAT

WAN Setting

IP Type: ☐ Fixed IP ☒ DHCP Client ☐ PPPoE

IP Address:

Mask:

Gateway:

DNS Server1:

DNS Server2:

MAC:

Host Name:

PPPoE Setting

User Name:

Password:

Service Name:

Lan Mode

If you set the LAN Mode as NAT (default) the device the ATA device will provide Network Address Translation between the WAN to LAN ports.

If you set the LAN Mode as Bridge, the device will by-pass all the packets from WAN to LAN directly.

WAN Setting

There are three modes available for the WAN IP address setting: Fixed IP, DHCP Client and PPPoE.

You can choose one to suit your situation.

If the WAN port is connected to a router, use DHCP (default) to get an address automatically, or Fixed IP to set an address manually.

If the WAN port is connected directly to a modem, you may need to use the PPPoE setting set up the connection to your ISP.

PPPoE Setting

If you set the WAN as PPPoE Client, you need to fill out the PPPoE user name and password given by your ISP.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.5.2 LAN Settings

LAN Settings

You can configure the LAN settings on this page.

| LAN Setting | |
|-------------|---------------|
| IP Address: | 192.168.3.1 |
| Mask: | 255.255.255.0 |
| MAC: | 0005b45000d5 |

| DHCP Server | |
|--------------|---|
| DHCP Server: | <input checked="" type="radio"/> On <input type="radio"/> Off |
| Start IP: | 150 |
| End IP: | 200 |
| Lease Time: | 1 : 0 (dd:hh) |

You can change the LAN IP address, and enable or disable the DHCP server on this page.

The default LAN IP address for the ATA device is 192.168.3.1 and DHCP is enabled to allow a PC or other device connected to the LAN port to obtain an IP address in the default range (.150 to .200) automatically on this subnet.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.5.3 STUN Settings

The screenshot shows the 'STUN Setting' page of the MVA11A web interface. On the left is a green sidebar with a menu of settings: Setup Wizard, System Status, Network (highlighted), WAN Settings, LAN Settings, STUN Settings, DDNS Settings, VLAN Settings, DMZ Setting, Virtual Server Settings, PPTP Settings, SIP Settings, Phone Book, Phone Settings, Advance, and System. The main content area is titled 'STUN Setting' and includes the instruction: 'You can set the IP of STUN server on this page.' Below this, there are three fields: 'STUN:' with radio buttons for 'On' and 'Off' (where 'Off' is selected), 'STUN Server:' with an empty text input box, and 'STUN Port:' with a text input box containing '3478' and a range '(80~65535)' in parentheses. At the bottom right of the form are 'Submit' and 'Reset' buttons.

You may setup a STUN server for VoIP Phone service if required.

This can be useful if operating behind a firewall but is otherwise not normally required and may be left set to Off.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.5.4 DDNS Settings

DDNS Settings

You can set the configuration of DDNS on this page..

DDNS: ☐ On ☒ Off

Host Name:

User Name:

Password:

E-mail Address:

DDNS Server:

DDNS Server List:

Type:

Wild Card:

BACKMX: ☐ On ☒ Off

Off Line: ☐ On ☒ Off

Dynamic DNS allows you to register and update a dynamic IP address with one of many available dynamic DNS service providers.

This allows Internet access to a FTP or Web service for example, running on your PC connected to the LAN port of the ATA, using DNS-like address.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.5.5 VLAN Settings

VLAN Settings

You can set the VLAN settings on this page.

| VLAN General Setting | |
|-------------------------|---|
| VLAN Packets: | <input type="radio"/> On <input checked="" type="radio"/> Off |
| VID (802.1Q/TAG): | 136 (2 ~ 4094) |
| User Priority (802.1P): | 1 (0 ~ 7) |
| CFI: | 1 (0 ~ 1) |

Submit Reset

You can enable the VLAN function at this page.

There are two parts in this page. First one is to set the packets related to the gateway, and the second part is if you use the VLAN setting in the NAT Mode.

There are two kinds of destination packets that will come from the ATA gateway's WAN port, one kind of packet will go to the ATA gateway, and the other will go through the LAN port to the PC.

-- **VLAN Packets:** if you enable the first VLAN Packets and set the VID, User Priority, and CFI, then all the incoming packets will be checked with the IP Address and the VID.

-- **VID:** Set as directed by your service provider.

-- **User Priority:** Defines user priority, giving eight (2^3) priority levels. IEEE 802.1P defines the operation for these three user priority bits. Usually this will be defined by your service provider.

-- **CFI:** Canonical Format Indicator is always set to zero for Ethernet switches. CFI is used for compatibility reason between Ethernet type network and Token Ring type network. If a frame received at an Ethernet port has a CFI set to 1, then that frame should not be forwarded as it is, to an untagged port.

When you enable the first VLAN Packets and set the VID, User Priority, and CFI, then all the incoming packets with the ATA gateway's IP address and the same VID will be accept by the ATA gateway. If the incoming packets with the ATA gateway's IP address but the different VID then the packets will be discarded by the ATA gateway.

Other incoming packets with different IP address will go through the LAN port to the PC.

NAT VLAN Setting: When you set your device in NAT mode, the ATA gateway can help you to filter unwanted incoming packets. You can separate the other devices connected behind the ATA gateway into four VLAN groups. You can set different VIDs for these four groups.

When the incoming packets go through the ATA gateway's WAN port then the Gateway will check the VID, if the packets are not going to the ATA gateway (with the ATA gateway's IP address and the correct VID), and the VID is not one of these four VIDs you have set, then the packets will be discarded by the ATA gateway.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

3.5.6 DMZ Settings

The screenshot shows the 'DMZ Setting' page. On the left is a green sidebar with a menu. The 'Network' section is highlighted in red. Under 'Network', 'DMZ Setting' is selected. Other options in the sidebar include 'Setup Wizard', 'System Status', 'WAN Settings', 'LAN Settings', 'STUN Settings', 'DDNS Settings', 'VLAN Settings', 'Virtual Server Settings', 'PPTP Settings', 'SIP Settings', 'Phone Book', 'Phone Settings', 'Advance', and 'System'. The main content area is titled 'DMZ Setting' and contains the text: 'You can configure your demilitarized zone setting on this page.' Below this, there are two settings: 'DMZ:' with radio buttons for 'On' and 'Off' (where 'Off' is selected), and 'DMZ Host IP:' with a text input field containing '0.0.0.0'. At the bottom right of the settings are 'Submit' and 'Reset' buttons.

You can set DMZ to mapping to the internal server or PC.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.5.7 Virtual Server Settings

Setup Wizard

System Status

Network

- WAN Settings
- LAN Settings
- STUN Settings
- DDNS Settings
- VLAN Settings
- DMZ Setting
- Virtual Server Settings
- PPTP Settings

SIP Settings

Phone Book

Phone Settings

Advance

System

Virtual Server Settings

You can set your virtual servers in this page. The usual port numbers are WEB [TCP 80], FTP(Control) [TCP 21], FTP(Data) [TCP 20], E-mail(POP3) [TCP 110], E-mail(SMTP) [TCP 25], DNS [UDP 53] and Telnet [TCP 23].

Virtual Server Page: Page 1 ▾

| Num | Enable | Protocol | In Port | Ex Port | Server IP | Select |
|-----|--------------------------|----------|---------|---------|-----------|--------------------------|
| 0 | <input type="checkbox"/> | | | | | <input type="checkbox"/> |
| 1 | <input type="checkbox"/> | | | | | <input type="checkbox"/> |
| 2 | <input type="checkbox"/> | | | | | <input type="checkbox"/> |
| 3 | <input type="checkbox"/> | | | | | <input type="checkbox"/> |
| 4 | <input type="checkbox"/> | | | | | <input type="checkbox"/> |
| 5 | <input type="checkbox"/> | | | | | <input type="checkbox"/> |
| 6 | <input type="checkbox"/> | | | | | <input type="checkbox"/> |
| 7 | <input type="checkbox"/> | | | | | <input type="checkbox"/> |

Enable Selected

Delete Selected

Delete All

Reset

You can setup Virtual Server definitions at this page and mapping to internal servers or PCs for special protocols.

The device supports a maximum of 24 virtual servers.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.5.8 PPTP Settings

The screenshot displays the PPTP Settings page. On the left is a sidebar with a green background containing several navigation buttons: Setup Wizard, System Status, Network (highlighted in red), WAN Settings, LAN Settings, STUN Settings, DDNS Settings, VLAN Settings, DMZ Setting, Virtual Server Settings, PPTP Settings, SIP Settings, Phone Book, Phone Settings, Advance, and System. The main content area is titled 'PPTP Settings' and includes the instruction 'You can set the PPTP server on this page.' Below this, there is a 'PPTP:' label followed by radio buttons for 'On' and 'Off', with 'Off' selected. Further down are three input fields labeled 'PPTP Server:', 'PPTP Username:', and 'PPTP Password:'. At the bottom right of the form are 'Submit' and 'Reset' buttons.

In this page, you can set the PPTP account details.

Fill out the PPTP Server, PPTP Account and Password here.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.6 SIP Settings

3.6.1 Service Domain

Service Domain Settings

You can set information of service domains on this page.

| Realm 1 (Default) | |
|--------------------|---|
| Active: | <input checked="" type="radio"/> On <input type="radio"/> Off |
| Display Name: | <input type="text"/> |
| User Name: | <input type="text"/> |
| Register Name: | <input type="text"/> |
| Register Password: | <input type="password"/> |
| Domain Server: | <input type="text"/> |
| Proxy Server: | <input type="text"/> |
| Outbound Proxy: | <input type="text"/> |
| Subscribe for MWI: | <input checked="" type="radio"/> On <input type="radio"/> Off |
| Status: | Not Registered |

| Realm 2 | |
|----------------|---|
| Active: | <input checked="" type="radio"/> On <input type="radio"/> Off |
| Display Name: | <input type="text"/> |
| User Name: | <input type="text"/> |
| Register Name: | <input type="text"/> |

In this page, you set up the SIP server settings. You need to input the account details as provided by your VSP or SIP provider. You can register up to three VSP/SIP accounts (Realms).

By default, outgoing calls will be made via the first account, and incoming may be received via all three accounts.

- First you need click on **Active** to enable the Realm dialog box, then you can input the following items:
- **Display Name**: input the name you want to display.
- **User Name**: input the User Name from your ISP.
- **Register Name**: input the Register Name from your ISP (often the same as User Name).
- **Register Password**: input the Register Password from your ISP.
- **Domain Server**: input the Domain Server from your ISP.
- **Proxy Server**: input the Proxy Server from your ISP (often the same as Domain Server).
- **Outbound Proxy**: input the Outbound Proxy get from your ISP. If your ISP does not provide the information, then you can leave this item blank.
- When you finished the setting, click the **Submit** button to apply the settings.
- You can see the Registration Status in the **Status** item. If the item shows "Registered", then your ATA Gateway has successfully registered to the ISP, you can make a phone call. If not, check the details for correctness.
- If you have more than one SIP account, you can follow the steps to register to the other accounts.

When you have finished the setting, click the Submit button.
The page will refresh, then click the Save button.

3.6.2 Port Settings

Port Settings

You can set the port number on this page.

| | | |
|------------------|-------|-------------------------|
| SIP Port: | 5060 | (0~65533, 0 for random) |
| RTP Port: | 20000 | (0~65533, 0 for random) |

You can setup the SIP and RTP port number in this page. VSP providers may have different SIP/ RTP port setting, please refer to the VSP information to setup the port number correctly.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.6.3 Codec Settings

| Codec Priority | |
|-------------------|-------------|
| Codec Priority 1: | G.711 u-law |
| Codec Priority 2: | G.711 a-law |
| Codec Priority 3: | G.723 |
| Codec Priority 4: | G.729 |
| Codec Priority 5: | G.726 - 16 |
| Codec Priority 6: | G.726 - 24 |
| Codec Priority 7: | G.726 - 32 |
| Codec Priority 8: | G.726 - 40 |
| Codec Priority 9: | GSM |

| RTP Packet Length | |
|-------------------|-------|
| G.711 & G.729: | 20 ms |
| G.723: | 30 ms |

| G.723 5.3K | |
|-------------|---|
| G.723 5.3K: | <input type="radio"/> On <input checked="" type="radio"/> Off |

On this page you can setup the Codec priority, RTP packet length, and VAD.

You need to follow the VSP / SIP provider's suggestion to setup these items. The default settings will not normally need to be changed.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.6.4 Codec ID Settings

Setup Wizard

System Status

Network

SIP Settings

- Service Domain
- Port Settings
- Codec Settings
- Codec ID Setting
- DTMF Setting
- RPort Setting
- Other Setting

Phone Book

Phone Settings

Advance

System

Codec ID Setting

You can set the value of Codec ID on this page. For most users, it is recommended to use default settings. If you want to change the values, please contact with your SIP provider.

| Codec Type | ID | Default Value |
|--------------|--------------|---|
| G726-16 ID: | 23 (95~255) | <input checked="" type="checkbox"/> 23 |
| G726-24 ID: | 22 (95~255) | <input checked="" type="checkbox"/> 22 |
| G726-32 ID: | 2 (95~255) | <input checked="" type="checkbox"/> 2 |
| G726-40 ID: | 21 (95~255) | <input checked="" type="checkbox"/> 21 |
| RFC 2833 ID: | 101 (95~255) | <input checked="" type="checkbox"/> 101 |

You may change the Codec IDs on this page.

Sometimes two VoIP devices with different Codec IDs on a SIP call will cause an interoperability issue which may be resolved by changing the Codec ID appropriately.

The default settings will not normally need to be changed.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.6.5 DTMF Settings

| DTMF Setting | |
|--|----------------------------------|
| You can set the DTMF setting on this page. | |
| RFC 2833 | <input checked="" type="radio"/> |
| Inband DTMF | <input type="radio"/> |
| Send DTMF SIP Info | <input type="radio"/> |
| <input type="button" value="Submit"/> <input type="button" value="Reset"/> | |

You can setup the RFC2833 Out-Band DTMF, In-band DTMF and Send DTMF SIP Info in this page.

To change this setting, please follow your ISP or SIP server provider information.

The default setting (RFC 2833) will not normally need to be changed.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.6.6 RPort Settings

RPort Setting

You can enable/disable the RPort setting on this page.

RPort: ☒ On ☐ Off

You can Enable/Disable the RPort in this page.

To change this setting, please follow your ISP or SIP server information.

The default setting (On) will not normally need to be changed.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.6.7 Other Settings

| Other Settings | |
|--|---|
| You can set other settings on this page. | |
| Hold by RFC: | <input type="radio"/> On <input checked="" type="radio"/> Off |
| Voice QoS (Diff-Serv): | 40 (0~63) |
| SIP QoS (Diff-Serv): | 40 (0~63) |
| SIP Expire Time: | 600 (15~86400 sec) |
| Use DNS SRV: | <input type="radio"/> On <input checked="" type="radio"/> Off |
| Send Keep Alives Packet: | <input checked="" type="radio"/> On <input type="radio"/> Off |
| Keep Alives Period: | 60 (15~250 sec) |
| Jitter Buffer: | 1 (0~250 packets) |

You can setup the Hold by RFC, Voice/SIP QoS and SIP expire time in this page.

To change these settings please following your ISP / VSP / SIP provider information.

The default settings will not normally need to be changed.

The QoS setting is to set the priority of voice packets. If you set the value higher than 0, then the voice packets will get the higher priority to the Internet, but the QoS function still has to cooperate with the other Internet devices.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.7 Phone Book Settings

3.7.1 Speed Dial Phone List

Phone Book

You can add/delete items in current phone book. Recommend to input digit number in name entries.

ex. Name: 201 , URL: 087065468. Speed dial press 201# then dial to number 087065468.

Phone Book Page: page 1

| Phone | Name | URL | Select |
|-------|------|-----|--------------------------|
| 0 | | | <input type="checkbox"/> |
| 1 | | | <input type="checkbox"/> |
| 2 | | | <input type="checkbox"/> |
| 3 | | | <input type="checkbox"/> |
| 4 | | | <input type="checkbox"/> |
| 5 | | | <input type="checkbox"/> |
| 6 | | | <input type="checkbox"/> |
| 7 | | | <input type="checkbox"/> |
| 8 | | | <input type="checkbox"/> |
| 9 | | | <input type="checkbox"/> |

Delete Selected Delete All Reset

The Phone Book List allows you to set up Speed Dial numbers.

To add a new number, scroll to the Add New Phone section of the page and enter the following data:

Position: The row number in the phonebook where the entry is to be stored (0 – 139)

Name: The sequence to be dialed to recall the required number: usually a number eg '201'

URL: The required phone number to be dialed, or SIP URI eg 12344567@sip.myvsp.com

When you have input the data, click the **"Add Phone"** button to save the entry into the device memory.

To clear the fields click the **"Reset"** button.

When you want to dial a number from the phonebook, dial the number string in the Name entry and then the # key eg "201#".

If you want to delete a phone number from the Phone book, you can select the phone number you want to delete then click **"Delete Selected"** button.

If you want to delete all phone numbers, you can click **"Delete All"** button.

Notes: When a user dials a number string followed by the # key, the device will search from the phone book first to see if the number matches a Phone Book Name entry.

If no match is found, it will dial the number directly.

Dial plan rules will be applied in dialing the numbers.

3.8 Phone Settings

3.8.1 Call Forward

Forward Setting

You can set the forward number of your phone on this page.

All Forward: ☒ Off ☐ IP ☐ PSTN

Busy Forward: ☒ Off ☐ IP

No Answer Forward: ☒ Off ☐ IP ☐ PSTN

| | Name | URL/Number |
|---------------------------|----------------------|----------------------|
| All Fwd No.: | <input type="text"/> | <input type="text"/> |
| Busy Fwd No.: | <input type="text"/> | <input type="text"/> |
| No Answer Fwd No.: | <input type="text"/> | <input type="text"/> |

No Answer Fwd Time Out: (2~8 Ring)

You can setup the Call Forward function in this page. There are three Call Forward modes as follows:

All Forward: All incoming calls will forward to the number you specify.

Busy Forward: If you are on the phone, a new incoming call will forward to the number you specify.

No Answer Forward: If you do not answer the phone, the incoming call will forward to the number you specify.

The top section of the screen is used to select whether each mode is active, and whether the outgoing call is to be placed via PSTN line or SIP.

The lower portion of the screen is used to specify the number to be called for each mode.

You can input a description in the **Name** field, and the Phone Number or SIP URI in **URL/Number** field. If the number matches an entry in the Phone Book list, then that number will be dialed.

The **Time Out** field specifies the number of rings for the system to before forwarding the call in the No Answer Forward mode.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.8.2 Volume Settings

| Volume Setting | | |
|--|---------------------------------|--|
| You can set the volume of your phone on this page. | | |
| Handset Volume: | <input type="text" value="10"/> | (0~12) |
| PSTN-Out Volume: | <input type="text" value="10"/> | (0~12) |
| Handset Gain: | <input type="text" value="10"/> | (0~15) |
| PSTN-In Gain: | <input type="text" value="10"/> | (0~15) |
| | | <input type="button" value="Submit"/> <input type="button" value="Reset"/> |

You can setup the Handset and PSTN Volume, and Handset and PSTN Gain.

Handset Volume: Set the receive volume for your handset.

PSTN-Out Volume: Set the PSTN receive volume.

Handset Gain: Set the send volume for your handset.

PSTN-In Gain: Set the PSTN send volume.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.8.3 DND Settings

DND Settings

You can set the do not disturb period of your phone on this page.

| | | |
|--------------------|---|--------------------------------------|
| DND Always: | <input type="radio"/> On | <input checked="" type="radio"/> Off |
| DND Period: | <input type="radio"/> On | <input checked="" type="radio"/> Off |
| From: | <input type="text" value="00"/> : <input type="text" value="00"/> | (hh:mm) |
| To: | <input type="text" value="00"/> : <input type="text" value="00"/> | (hh:mm) |

You can setup the DND (Do Not Disturb) setting to keep the phone from ringing.

There are two DND modes of operation as follows:

DND Always: All incoming calls will be blocked until you disable this feature.

DND Period: Set a time of day period for the phone to be blocked to incoming calls.
If the "From" time is large than the "To" time, the Block time will extend from Day 1 to Day 2.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.8.4 Auto Answer

Auto Answer

You can enable/disable the auto answer on this page.

| | |
|-----------------------------|--|
| Auto Answer: | <input checked="" type="radio"/> Off <input type="radio"/> IP In <input type="radio"/> FXO In <input type="radio"/> Both <input type="radio"/> Trunk Gateway |
| Auto Answer Counter: | <input type="text" value="3"/> (0~8) |
| PIN Code Enabled: | <input checked="" type="radio"/> Off <input type="radio"/> On |
| PIN Code: | <input type="text"/> |

This screen allows you to set up the Auto Answer function that enables you to call in to the device with PSTN and call out on SIP, or call in on SIP and call out on PSTN.

Auto Answer: If set to “On”, the ATA device will answer incoming calls.

Note: The Trunk Gateway feature requires support from your SIP service provider.

Auto Answer Counter: Sets the number of ring counts before the device answers the call.

Pin Code Enabled: For security it is advised that you set this to “On”

Note: If Trunk Gateway mode is enabled, the PIN Code should **not** be enabled.

PIN Code: Enter the PIN code you wish to use to allow the outgoing calls.

Operation

If a call comes in to the device and is not answered, then after the specified ring count is reached, the device will answer the call and send a series of short tones while waiting for the caller to enter the PIN code.

The PIN code is entered from the callers phone pad, and followed by the # key.

The device will then present dial tone for the outgoing call.

If the PIN code is not enabled, dial tone will be presented immediately on answer.

If the call has come in from the SIP port, then the outgoing call will be made on the PSTN line.

If the call has come from the PSTN line, then the outgoing call will be made on the SIP port.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot.

3.8.5 Caller ID

| Caller ID Setting | |
|--|---|
| You can enable/disable the caller ID setting on this page. | |
| Caller ID: | Caller ID after 1st Ring (FSK) ▼ |
| Single Caller ID: | <input type="radio"/> Yes <input checked="" type="radio"/> No |
| CID Without Time: | <input type="radio"/> Yes <input checked="" type="radio"/> No |
| CID Type 2: | <input type="radio"/> Yes <input checked="" type="radio"/> No |
| <input type="button" value="Submit"/> <input type="button" value="Reset"/> | |

On this screen you can set the ATA device to send Caller ID for outgoing PSTN or SIP calls in the method appropriate to your local telco standards.

The Caller Id may be Disabled or set to one of three types.

The correct setting will depend on the telco standards in operation in your area.

To prevent Caller ID being sent select ' Don't show Caller ID' from the drop down list.

The default settings will not normally need to be changed.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.8.6 Dial Plan Settings

Dial Plan

You can the set the dial plan on this page.

Routing to : ☐ IP ☐ FXO ☒ Disable

Routing rule :

Drop prefix : ☐ Yes ☒ No

Replace rule 1: +

Drop prefix: ☐ Yes ☒ No

Replace rule 2: +

Drop prefix: ☐ Yes ☒ No

Replace rule 3: +

Drop prefix: ☐ Yes ☒ No

Replace rule 4: +

Dial now:

Auto Dial Time: 5 (3~9 sec)

Use # as send key: ☒ Yes ☐ No

Use * for IP dialing: ☒ Yes ☐ No

Use PSTN backup: ☐ Yes ☒ No

Dial Plan rules allow you to change the way the device makes outgoing calls, for example adding the area code for local calls.

Prefix Handling

You can define up to four Dial Plan rules involving Prefix handling.

The Prefix is just the first part of a dialed number.

The dial plan rules are based on matching the patterns of dialed digits. When a pattern is matched the rule is applied.

Drop Prefix: Select whether the Prefix is to be dialed or not if the rule is activated.

Replace rule: The Prefix Pattern consists of two fields. Operation of the prefix handling rule is as follows:

- If the Drop Prefix is set to "No", and the number dialed matches the second Prefix Pattern field, then the number will be sent out with the first Prefix field added at the start.
- If the Drop Prefix is set to "Yes", and the number dialed matches the second Prefix Pattern field, then the number will be sent out with the first Prefix field replacing the dialed Prefix, i.e. the dialed Prefix digits are replaced by the first Prefix field.

Dial Now: If this pattern is matched, the number will be sent out immediately.

Auto Dial Time: When this time period is reached after the last digit is dialed, the dialed number is sent out i.e. if the dialed number string does not match any other rule, it will be sent out after this time period.

Use # as send key: If this is set to Yes, dialing # character after the dialed number will cause the number to be sent out immediately.

Use * for IP dialing: This setting allows the use of the * character as a delimiter in IP addresses used for direct IP dialing. For example:
192.168.1.100:5061 may be dialed as 192*168*1*100**5061

When you have finished the setting, click the Submit button.
The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

Dial Plan Syntax

- 0-9** - Treated as normal digits
- x, X** - Treated as any digit in the range 0-9
- *** - Treated as normal * character
- #** - Treated as normal # character

Digits in a pattern are represented by the character 'x' or 'X', ie 'X' means any digit 0 - 9.

| | |
|--------|---------------------|
| x or X | 0,1,2,3,4,5,6,7,8,9 |
| + | OR |

Multiple Patterns in a rule

Within each rule, a number of patterns may be specified by separating them with the '+' character. This acts as an OR operator ie if <Pattern A> OR <Pattern B> is matched, then the rule is applied.

Example: 3XXXXXXX+4XXXXXXX+5XXXXXXX

This rule will match an eight digit number which begins with 3 or 4 or 5.

Example Dial Plan Prefix Handling Rules

| Drop Prefix | Prefix | Pattern | Rule |
|-------------|--------|-----------|--|
| No | 07 | 3+4+5 | If the dialed number starts with 3 or 4 or 5, the number is sent out with 07 added to the beginning. (Used for adding local area codes in Qld) |
| No | 03 | 4+5+6+8+9 | If the dialed number starts with 4,5,6,8 or 9, the number is sent out with 03 added to the beginning. (Used for adding local area codes in Vic / Tas) |
| No | 02 | 4+5+6+8+9 | If the dialed number starts with 4,5,6,8 or 9, the number is sent out with 02 added to the beginning. (Used for adding local area codes in NSW) |
| No | 08 | 6+7+8+9 | If the dialed number starts with 6,7,8 or 9, the number is sent out with 08 added to the beginning. (Used for adding local area codes in S. Aust / W. Aust / North. Territory) |
| Yes | 006 | 002 + 003 | If a prefix of either 002 or 003 is dialed, it will be replaced by 006. |
| Yes | 07 | * | If a * character is dialed at the beginning of a number, it will be replaced by 07. |

3.8.7 Flash Time Settings

Flash Time Setting

You can set the flash time on this page.

FXO Flash Time

Flash Time: 60 x 10 ms (9~120)

FXS Flash Time

Max Flash Time: 60 x 10 ms (4~255)

Min Flash Time: 3 x 10 ms (3~12)

Submit Reset

This screen allows you to set the timing parameters for Hook / Flash (a momentary break in the phone line circuit) to activate certain functions such as switching to the other line or placing the line on hold.

The timing parameters are set independently for the FXO and FXS ports.

The default setting will not normally need to be varied.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.8.8 Call Waiting Settings

Call Waiting Setting

You can enable/disable the call waiting setting on this page.

Call Waiting: ☒ On ☐ Off

You can Enable/Disable the Call Waiting function in this screen.

With **Call Waiting** activated, when you are talking with someone on a call and there is a new incoming call, you will hear the call waiting tone.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.8.9 T.38 (FAX) Settings

T.38 (FAX) Setting

You can enable/disable the FAX function on this page.

| | |
|--------------------|---|
| T.38 (FAX): | <input checked="" type="radio"/> On <input type="radio"/> Off |
| T.38 Port: | <input type="text" value="20000"/> (1024~65533) |

You can Enable/Disable the T.38 function and set the T.38 Port number on this screen.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.8.10 Hot line Settings

The screenshot shows a web interface for 'Hot line Setting'. On the left is a green sidebar with a list of settings: Setup Wizard, System Status, Network, SIP Settings, Phone Book, Phone Settings (highlighted in red), and sub-items under Phone Settings: Call Forward, Volume Settings, DND Settings, Auto Answer, Caller ID, Dial Plan Setting, Flash Time Setting, Call Waiting Setting, T.38 (FAX) Setting, Hot line Setting, and Alarm Settings. Below these are 'Advance' and 'System' buttons. The main content area is titled 'Hot line Setting' and contains the text 'You can set the hot line on this page.' Below this are two labels: 'Use Hot Line:' and 'Hot line number:'. The 'Use Hot Line:' label is next to radio buttons for 'Enable' and 'Disable', with 'Disable' selected. The 'Hot line number:' label is next to a text input field. At the bottom right are 'Submit' and 'Reset' buttons.

You can Enable/Disable the Hot line function and set the Hot Line number on this screen.

If this function is enabled, when you pick up the phone, it will automatically dial the Hot Line number you have set up.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.8.11 Alarm Settings

Alarm Settings

You can set the alarm time on this page.

| | |
|----------------------|---|
| Alarm: | <input type="radio"/> On <input checked="" type="radio"/> Off |
| Alarm Time: | <input type="text" value="0"/> : <input type="text" value="0"/> (hh:mm) |
| Current time: | 2005-01-01 09:04 |

You can set the time to alarm via the VoIP ATA.

With the Alarm set to ON, the ATA will ring the connected phone at the Alarm Time.

The Current Time is shown for reference.

Check that the Current Time is shown correctly. If it is not shown correctly, check the settings in the Network / WAN menu and the Advance / SNTP Settings menu to ensure that the device can access an Internet time service, and that the Time Zone is set correctly for your local area.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.9 Advance Settings

3.9.1 Auto Config

Auto Configuration Setting

You can enable/disable the auto configuration setting on this page.

Auto Configuration: ☒ Off ☐ TFTP ☐ FTP ☐ HTTP

TFTP Server:

HTTP Server:

HTTP Path:

FTP Server:

FTP Username:

FTP Password:

File Path:

This function will automatically download the configuration file to setup your Gateway.

Auto Configuration: Select the type of protocol to be used, or Off to disable the function.

TFTP/HTTP/FTP Server: Set up the relevant server addresses, paths and login details for the protocols you wish to use.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.9.2 SNTP Settings

SNTP Settings

You can set the SNTP servers on this page.

| | |
|--------------------------|---|
| SNTP: | <input checked="" type="radio"/> On <input type="radio"/> Off |
| Primary Server: | <input type="text" value="time.windows.com"/> |
| Secondary Server: | <input type="text" value="208.184.49.9"/> |
| Time Zone: | GMT + 08 : 00 (hh:mm) |
| Sync. Time: | 1 : 0 : 0 (dd:hh:mm) |
| Current time: | 2005-01-01 09:12 |

This screen allows you to set up time synchronization for the device.

SNTP: Set to On to allow the ATA clock to synchronise with an internet source.

Primary/Secondary Server: Set up the primary and second SNTP Server IP Addresses to get the date/time information to set the ATA clock.

Time Zone: Set the Time Zone to match your location so that the clock is set to local time.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.9.3 FXO & FXS Port

FXO & FXS Impedance Setting

You can select the FXO & FXS impedance of the analog telephone by different country on this page.

FXO Port: Australia ▼

FXS Port: Australia ▼

FXO Silence Timeout : 100 (1~250 minutes)

You can set up the FXO or FXS line impedance in this page.

Select the impedance setting for the FXO and FXS ports to match the line impedance of your local PSTN system.

The settings shown above are the default settings and will not normally need to be changed.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.9.4 Tones Settings

- [Setup Wizard](#)
- [System Status](#)
- [Network](#)
- [SIP Settings](#)
- [Phone Book](#)
- [Phone Settings](#)
- [Advance](#)
- Auto Provision
- SNTP Settings
- FXO & FXS Port
- Tones Settings
- Advanced Setting
- MAC Clone Setting
- [System](#)

Tones Settings

You can configure your tones settings on this page.

| | Dial Tone | Ring Back Tone | Busy Tone | Error Tone | Ring Tone | Insert Tone |
|----------------|-----------------------------------|-------------------------------------|-------------------------------------|-------------------------------------|-------------------------------------|-------------------------------------|
| Cadence On: | <input type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> |
| Hi-Tone Freq.: | <input type="text" value="440"/> | <input type="text" value="480"/> | <input type="text" value="620"/> | <input type="text" value="620"/> | <input type="text" value="480"/> | <input type="text" value="440"/> |
| Lo-Tone Freq.: | <input type="text" value="350"/> | <input type="text" value="440"/> | <input type="text" value="480"/> | <input type="text" value="480"/> | <input type="text" value="440"/> | <input type="text" value="350"/> |
| Hi-Tone Gain: | <input type="text" value="4522"/> | <input type="text" value="2261"/> | <input type="text" value="2261"/> | <input type="text" value="2261"/> | <input type="text" value="15360"/> | <input type="text" value="2261"/> |
| Lo-Tone Gain: | <input type="text" value="2261"/> | <input type="text" value="2261"/> | <input type="text" value="2261"/> | <input type="text" value="2261"/> | <input type="text" value="15360"/> | <input type="text" value="1130"/> |
| On Time 1: | <input type="text" value="0"/> | <input type="text" value="200"/> | <input type="text" value="50"/> | <input type="text" value="30"/> | <input type="text" value="200"/> | <input type="text" value="30"/> |
| Off Time 1: | <input type="text" value="0"/> | <input type="text" value="400"/> | <input type="text" value="50"/> | <input type="text" value="20"/> | <input type="text" value="400"/> | <input type="text" value="20"/> |
| On Time 2: | <input type="text" value="0"/> | <input type="text" value="0"/> | <input type="text" value="0"/> | <input type="text" value="0"/> | <input type="text" value="0"/> | <input type="text" value="30"/> |
| Off Time 2: | <input type="text" value="0"/> | <input type="text" value="0"/> | <input type="text" value="0"/> | <input type="text" value="0"/> | <input type="text" value="0"/> | <input type="text" value="400"/> |
| On Time 3: | <input type="text" value="0"/> | <input type="text" value="0"/> | <input type="text" value="0"/> | <input type="text" value="0"/> | <input type="text" value="0"/> | <input type="text" value="0"/> |
| Off Time 3: | <input type="text" value="0"/> | <input type="text" value="0"/> | <input type="text" value="0"/> | <input type="text" value="0"/> | <input type="text" value="0"/> | <input type="text" value="0"/> |

You can set the VoIP Tone parameters in this page.

You can set the Dial tone, Ring Back tone, Busy tone, Error tone, Ring tone and Insert tone. The value for each column can be 0 ~ 99999.

Check with your VSP for the detailed values for the tones if required.

The default settings will not normally need to be changed.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.9.5 Advanced Settings

You can setup advanced functions for the device in this page.

Send Anonymous CID: If enabled, this will hide the phone number from the send side; the receiver will not show the phone number.

Management from WAN: Allows the web administration interface to be accessed from the Internet. Caution: This can pose a security threat. Disable if not specifically required.

Billing Signal: Controls a polarity reversal pulse to support certain telco billing systems if required.

CPC Delay and Duration: May be used to control telco line release on hang up.

Send Flash Event: Controls sending of hook / flash for telco functions.

SIP Encrypt: Allows you to select SIP encryption. Default is disabled.

PPPoE retry period: Controls PPPoE reconnect timing.

System Log Server: Specify the address of a syslog server.

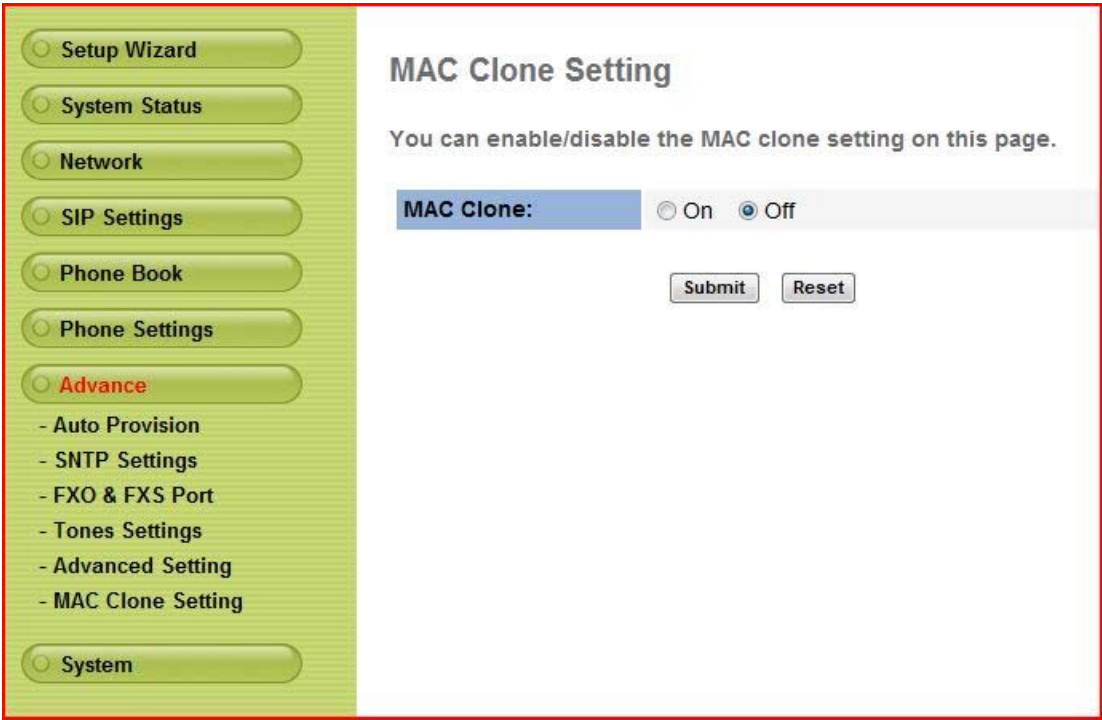
System Log Type: Select the type of events to be logged.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.9.6 MAC Clone Settings



The screenshot shows a web interface for configuring MAC Clone settings. On the left is a green sidebar with a list of settings: Setup Wizard, System Status, Network, SIP Settings, Phone Book, Phone Settings, Advance (highlighted in red), and System. Under the 'Advance' section, there are sub-items: Auto Provision, SNTP Settings, FXO & FXS Port, Tones Settings, Advanced Setting, and MAC Clone Setting. The main content area is titled 'MAC Clone Setting' and contains the text: 'You can enable/disable the MAC clone setting on this page.' Below this text is a label 'MAC Clone:' followed by two radio buttons: 'On' and 'Off'. The 'Off' radio button is selected. At the bottom of the main area are two buttons: 'Submit' and 'Reset'.

You can set up the MAC Clone Enable/Disable in this page.

If enabled, this function will automatically clone the connected PC's LAN card MAC address to the WAN port to support some ISP arrangements to lock the PPPoE client's MAC address.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.10 System Settings

3.10.1 System Authority

The screenshot shows the 'System Authority' settings page. On the left is a green sidebar with a list of settings categories: Setup Wizard, System Status, Network, SIP Settings, Phone Book, Phone Settings, Advance, and System. The 'System' category is selected and highlighted in red. Under 'System', there are sub-items: System Authority, Save Changes, Upgrade Firmware, Default Settings, Backup Settings, and Reboot System. The main content area is titled 'System Authority' and contains the text 'You can change the login username/password on this page.' Below this text are three input fields: 'New username:', 'New password:', and 'Confirmed password:'. Each field has a blue label and a white input box. At the bottom right of the form are two buttons: 'Submit' (blue) and 'Reset' (grey).

You can change your login name and password on this screen.

It is recommended that you change the default Admin password to ensure security of your system.

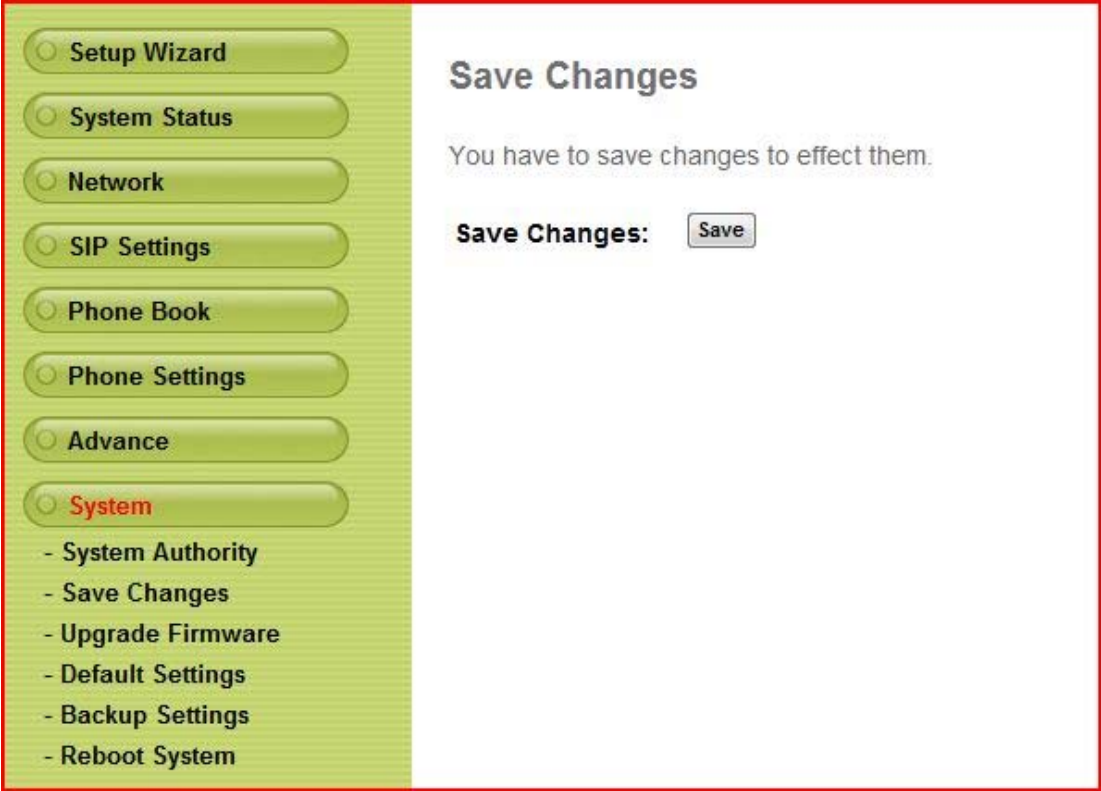
Note: A Factory Settings reset will restore the default Admin password in case you forget your password.

When you have finished the setting, click the Submit button.

The page will refresh, then click the Save button.

The changes you made will be saved into the system memory and the system will reboot automatically.

3.10.2 Save Changes



Save Changes

You have to save changes to effect them.

Save Changes:

You can save the setting changes you have made into system memory from this page.

The system will automatically reboot after saving the configuration settings.

3.10.3 Update Firmware

You can update the ATA Gateway's firmware or do a "factory reset" to set the device back to default settings.

Select the **Method** you wish to use to update new firmware.

You can select Local PC to use a file saved in your file system to be uploaded by the browser using HTTP.

Alternatively you can select TFTP to use a Trivial File Transfer Protocol server to provide the required file.

Upgrade the firmware by the following steps:

Select the firmware code type, **Risc** or **DSP** code according to which module you wish to update.

For **Local PC** mode, click the "Browse" button in the right side of the File Location to find the required file, or you can type the correct path and the filename in File Location blank.

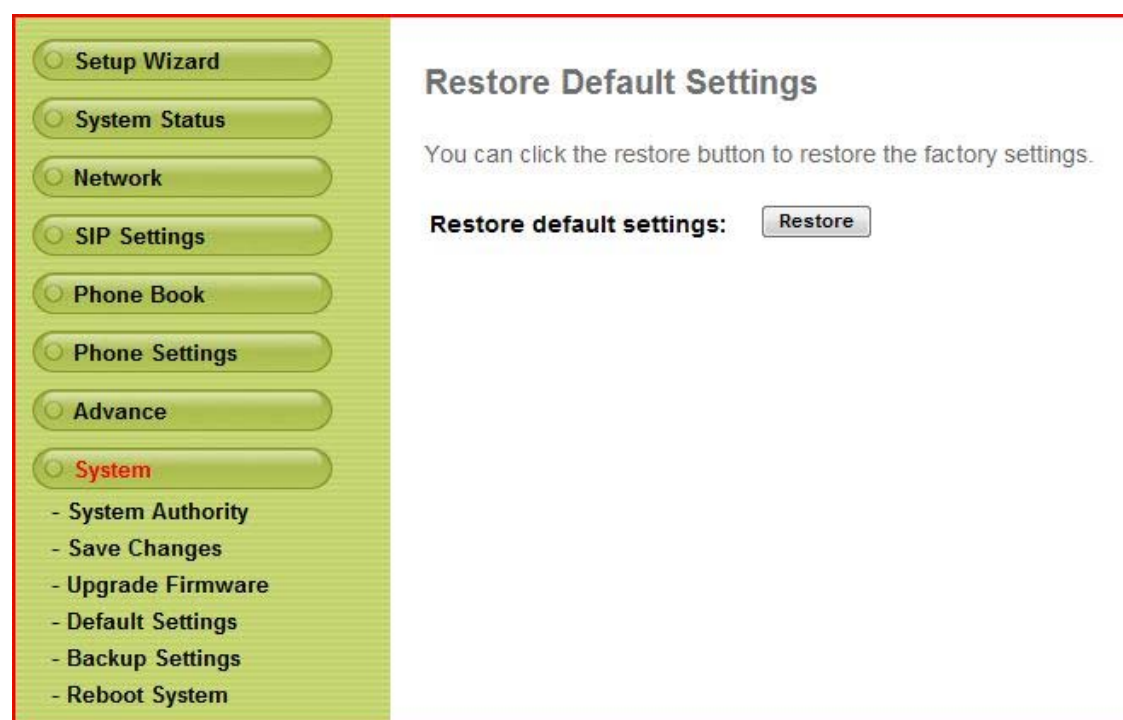
For **TFTP** mode, specify the TFTP server IP Address

Then click the **Update** button.

Caution: Do not disturb the device, power or network connections while the upgrade process is in operation as this may corrupt the transfer and render the device unusable.

After the firmware update process has finished, the system will reboot automatically.

3.10.4 Default Settings



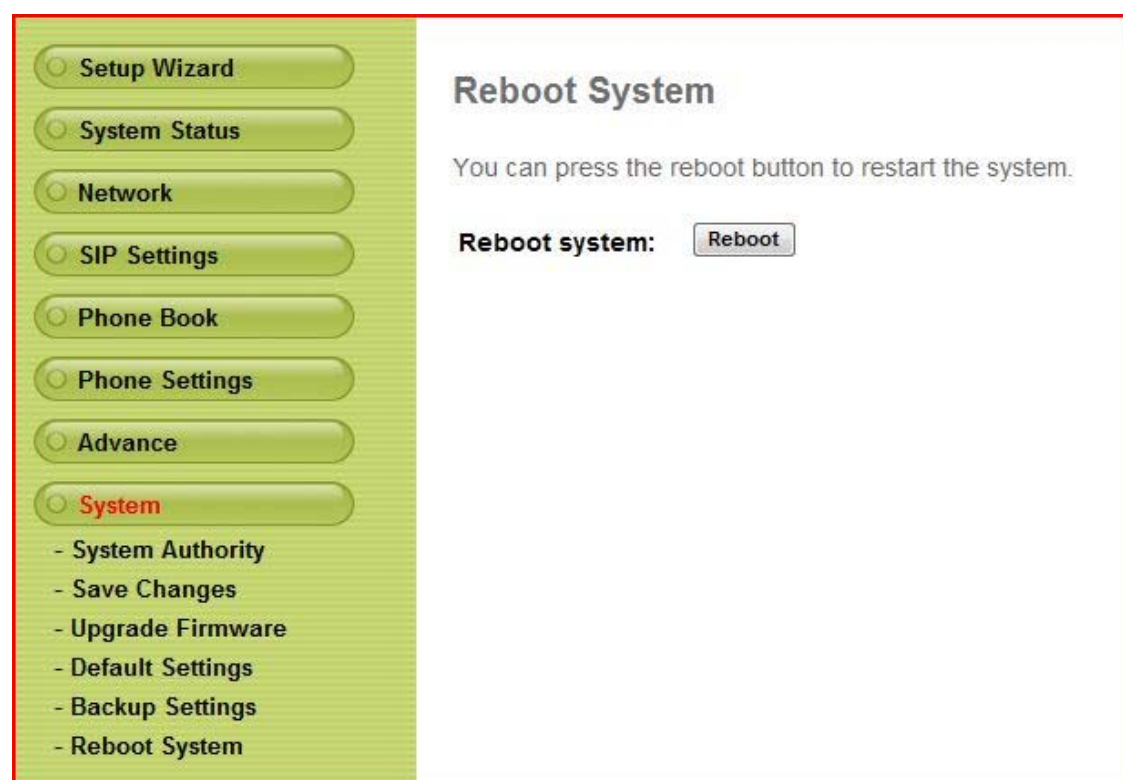
You can restore the device to factory default in this page.

Click the Restore button, and the device will restore to default settings and automatically restart.

The Default Settings will be:

- NAT Mode,
- WAN port is set as DHCP Client Mode,
- LAN port is Fixed IP Mode and the IP Address is 192.168.3.1.

3.10.5 Reboot System



You can restart the device from this page.

If you want to restart the device, just click the Reboot button, and then the device will reboot automatically.

3.11 Interactive Voice Response (IVR) interface for the Gateway

You can use the PSTN phone to query and configure the device using Tone Dial pad.
Please follow the instructions below to query / configure your terminal adapter.

| Group | IVR Action | IVR Menu Choice | Parameter(s) | Notes |
|----------|----------------------------------|--------------------------|--|--|
| Function | enable call waiting | #138# | None | Enable Call waiting |
| Function | disable call waiting | #139# | None | Disable call waiting |
| Function | unlock keypad | #190# | None | You have to unlock keypad first, and then you can change the setting by keypad. |
| Function | lock keypad | #191# | None | Lock keypad. |
| Function | Reboot | #195# | None | The system will reboot automatically. |
| Function | Factory Reset | #198# | None | System will automatically Reboot and restore to default setting. WARNING: ALL "User-Changeable" NONDEFAULT SETTINGS WILL BE LOST! This will include network and service provider data. |
| Info | Check IP Address | #120# | None | IVR will report the LAN port IP address |
| Info | Check IP Type | #121# | None | IVR will report the WAN Port DHCP is enabled or disabled. |
| Info | Check the Phone Number | #122# | None | IVR will report current in use VoIP number |
| Info | Check Network Mask | #123# | None | IVR will report the WAN Port network mask |
| Info | Check Gateway IP Address | #124# | None | IVR will report the WAN Port gateway IP address |
| Info | Check Primary DNS Server Setting | #125# | None | IVR will report the WAN Port Primary DNS server IP Address. |
| Info | Check IP Address | #126# | None | IVR will report the WAN port IP address |
| Info | Check Firmware Version | #128# | None | IVR will report the firmware version |
| Setting | Set DHCP client | #111# | None | The system will change the WAN port to DHCP Client type |
| Setting | Set Static IP Address | #112xxx*xx x*xxx*xxx# | Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point. | WAN port DHCP Client will be disabled and WAN port will change to the Static IP type. Set WAN port IP Address |
| Setting | Set Network Mask | #113xxx*xx x*xxx*xxx# | Enter value using numbers on the telephone key pad. Use the * (star) key when entering a decimal point. | Must set Static IP first. Set WAN port Network Mask |
| Setting | Set Gateway IP Address | #114xxx*xx x*xxx*xxx# | Enter IP address using numbers on | Must set Static IP first. Set WAN port Gateway IP |

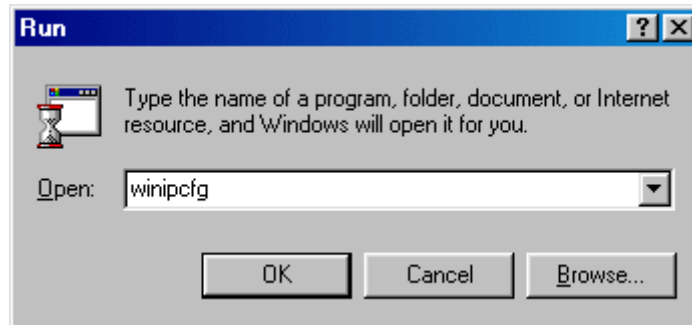
| | | | | |
|---------|------------------------|-----------------------------------|---|--|
| | | | the telephone key pad. Use the * (star) key when entering a decimal point. | Address |
| Setting | Set Primary DNS Server | #115xxx*xx x*xxx*xxx# | Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point. | Must set Static IP first. Set WAN port Primary DNS Server IP Address |
| Setting | Set Codec | #130+[1-8]# | 1:G.711 u-Law, 2: G.711 a-Law, 3: G.723.1, 4: G.729a, 5: G.726 16K, 6: G.726 24K, 7: G.726 32K, 8: G.726 40K, | You can set the codec you want to the first priority. |
| Setting | Set Handset Gain | #131+[00~15]# | Handset Gain from 0~15 | You can set the Handset gain to proper value, default is 10 |
| Setting | Set Handset Volume | #132+[00~12]# | Handset Volume from 0~12 | You can set the Handset volume to proper value, default is 10 |
| Setting | TFTP Server IP Address | #135# | Set Auto configure TFTP Server IP Address | You can set the TFTP Server IP address |
| Setting | FTP Server IP Address | #136# | Set Auto configure FTP Server IP Address | You can set the FTP Server IP address |
| Setting | Auto configure mode | #137+[0~2]# | 0: Disable, 1: TFTP mode, 2: FTP mode | You can set the Auto configuration mode, 0: Disable, 1: use TFTP Server, 2: user FTP Server |
| Setting | Blind transfer | flash#510# Phone no.# | None | B calls A, and A transfers the call to C. A needs to press flash button and then press #510#(phone no. of C)#, then A hangs up the phone. Then the call connects B and C. |
| Setting | Attendant transfer | flash#511# Phone no.# | None | B calls A, and A transfers the call to C. A needs to press flash button and then press #511#(phone no. of C)#. The call establishes between A and C. When A hangs up the phone, the call connects B and C. |
| Setting | 3-way-calling | flash#512# Phone no.# flash | None | B calls A, and A conferences the call to C. A needs to press flash button and then press #512#(phone no. of C)#. When C picks up the phone, A needs to press flash again, and then begin 3-way conference. |
| Setting | PSTN mode | 0* | None | Set default setting to PSTN mode |
| Setting | Realm Switch | 1* 2* 3* | None | Set the realm for an outgoing call. |

3.12 TCP/IP Settings for Windows Operating System

1. How can I find my IP Address in Windows 95, 98, or Me?

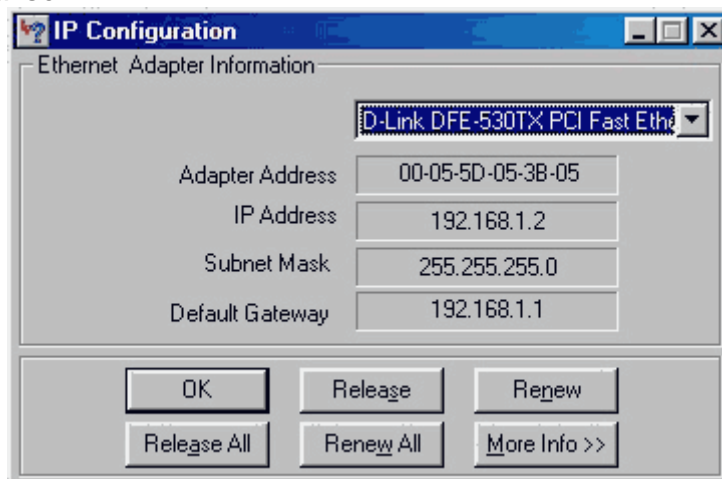
Click on Start, then click on Run.

The Run Dialogue Box will appear. Type winipcfg in the window as shown then click OK



The **IP Configuration** window will appear, displaying your **Ethernet Adapter Information**. Select your adapter from the drop down menu.

If you do not see your adapter in the drop down menu, your adapter is not properly installed.



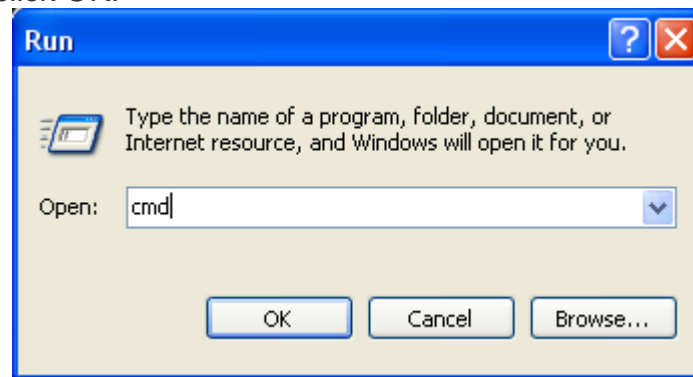
After selecting your adapter, it will display your IP Address, subnet mask, and default gateway.

Click OK to close the IP Configuration window.

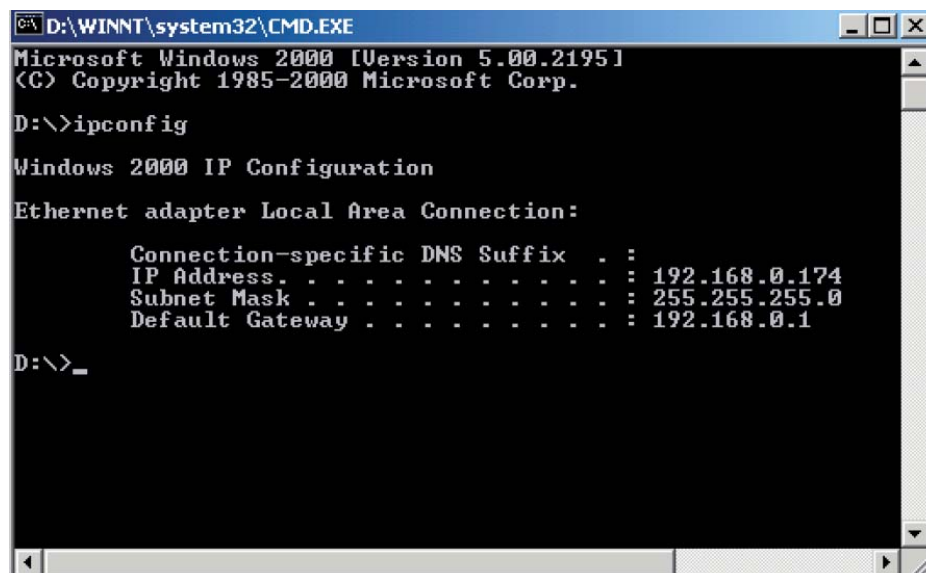
2. How can I find my IP Address in Windows 2000/XP?

Click on Start and select Run.

Type cmd then click OK.



From the Command Prompt, enter ipconfig. It will return your IP Address, subnet mask, and default gateway.



Type exit to close the command prompt.

Make sure you take note of your computer's Default Gateway IP Address.

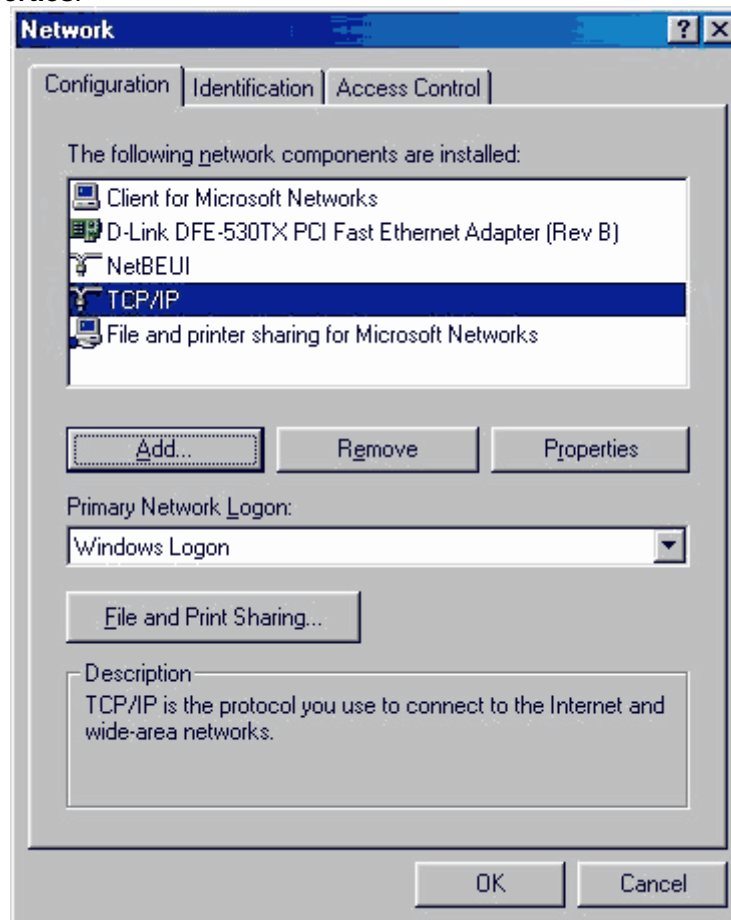
The Default Gateway is the IP Address of the router.

By default, it should be 192.168.0.1

3. How can I assign a Static IP Address in Windows 98/Me?

From the desktop, right-click on the **Network Neighborhood** icon (Win ME - My Network Places) and select **Properties**.

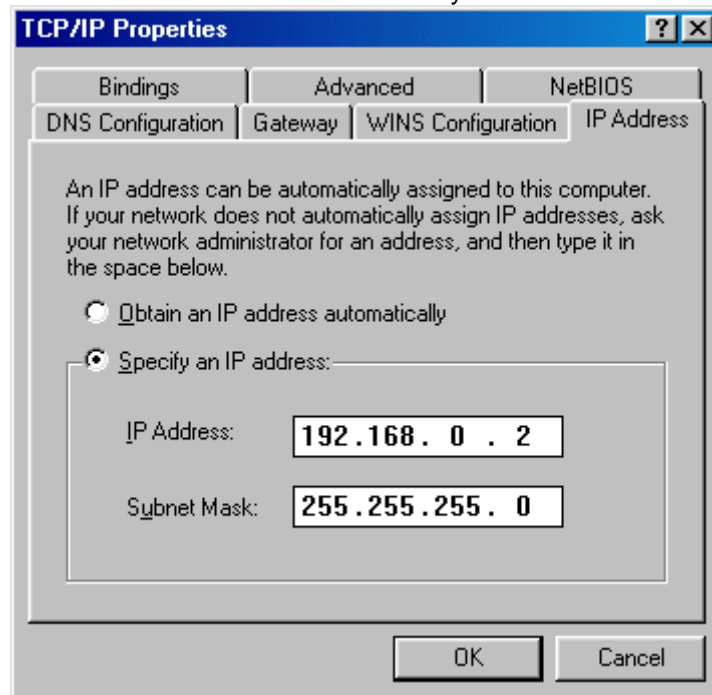
Highlight **TCP/IP** and click the **Properties** button. If you have more than 1 adapter, then there will be a TCP/IP "Binding" for each adapter. Highlight **TCP/IP > (your network adapter)** and then click **Properties**.



Click **Specify an IP Address**.

Enter an IP Address that is on the same subnet as the LAN IP Address on your router.

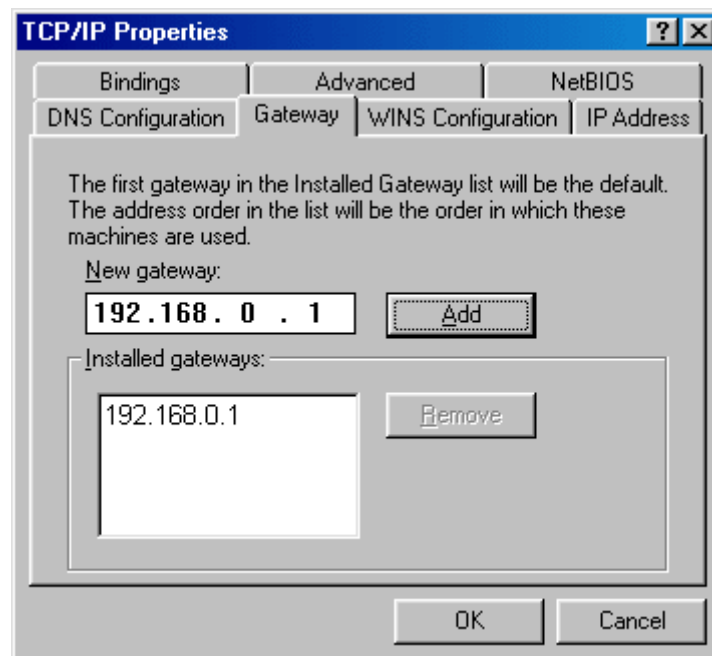
Example: If the router's LAN IP Address is 192.168.0.1, make your IP Address 192.168.0.X where X is between 2-99. Make sure that the number you choose is not in use on the network.



Click on the **Gateway** tab.

Enter the LAN IP Address of your router here (192.168.0.1).

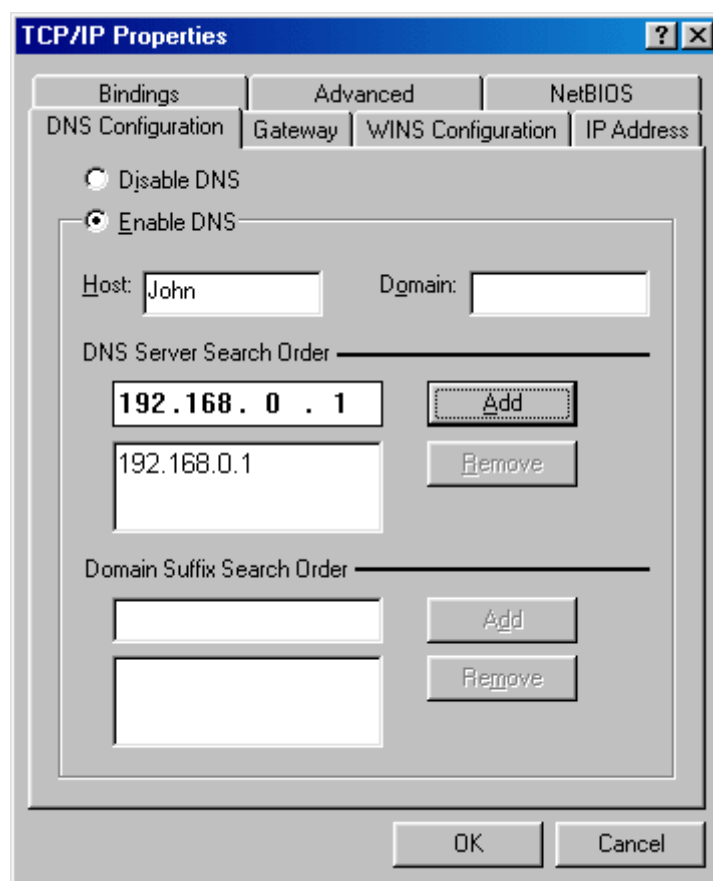
Click **Add** when finished.



Click on the **DNS Configuration** tab.

Click **Enable DNS**. Type in a **Host** (can be any word). Under DNS server search order, enter the LAN IP Address of your router (192.168.0.1).

Click **Add**.



Click **OK** twice.

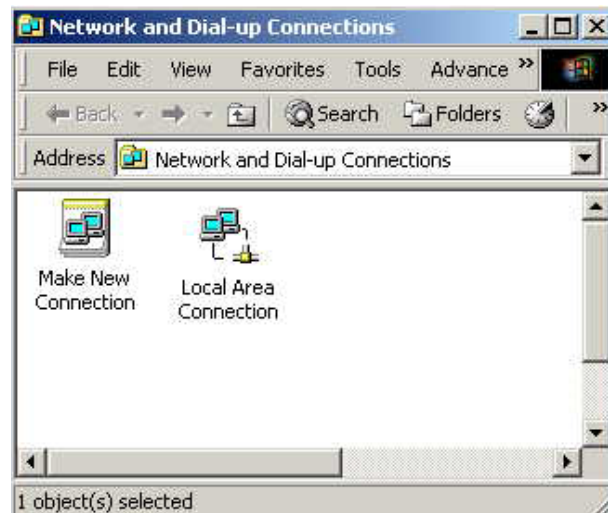
When prompted to reboot your computer, click **Yes**.

After you reboot, the computer will now have a static, private IP Address.

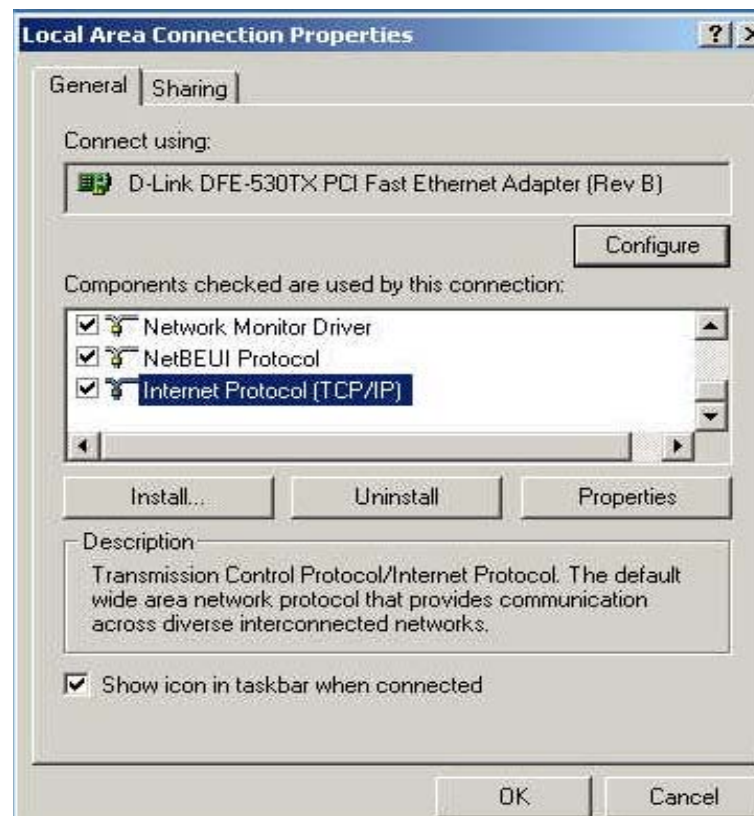
4. How can I assign a Static IP Address in Windows 2000?

Right-click on My Network Places and select Properties.

Right-click on the Local Area Connection which represents your network card and select Properties.



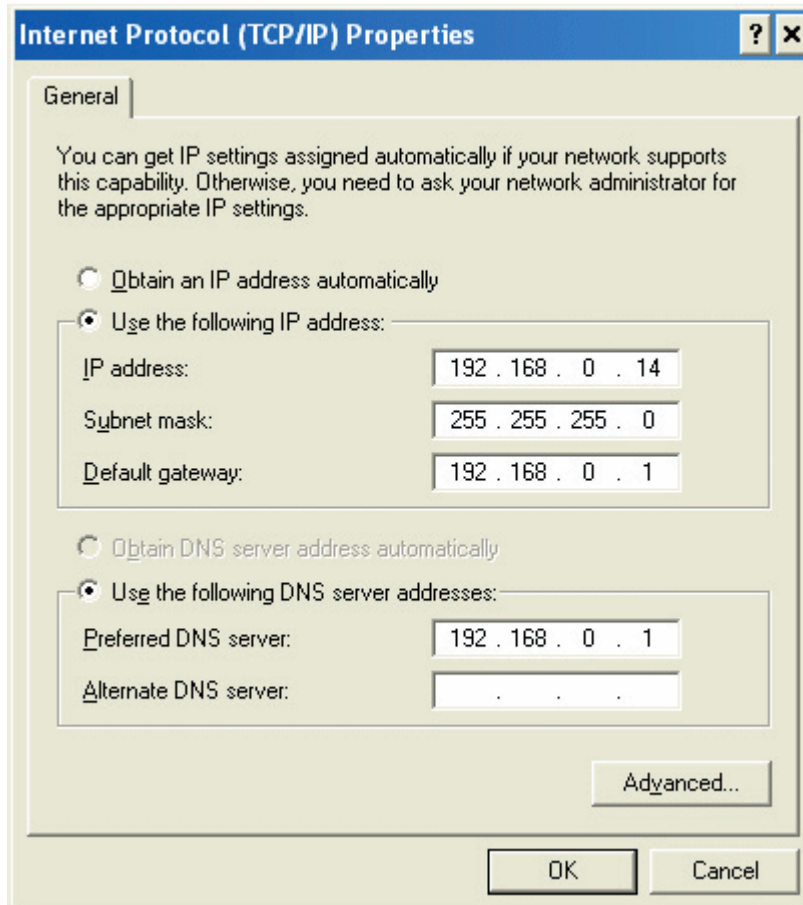
Highlight Internet Protocol (TCP/IP) and click Properties.



Click **Use the following IP Address** and enter an IP Address that is on the same subnet as the LAN IP Address on your router. Example: If the router's LAN IP Address is 192.168.0.1, make your IP Address 192.168.0.X where X = 2-99. Make sure that the number you choose is not in use on the network. Set the **Default Gateway** to be the same as the LAN IP Address of your router (192.168.0.1). Set the **Primary DNS** to be the same as the LAN IP address of your router (192.168.0.1). The **Secondary DNS** is not needed, or you may enter a DNS server from your ISP. Click **OK** twice. User may be asked if you want to reboot your computer. Click **Yes**.

5. How can I assign a Static IP Address in Windows XP?

Click on Start > Control Panel > Network and Internet Connections > Network connections. See the steps for assigning a static IP address in Windows 2000 and continue from there.



Access the Web management. Open your Web browser and enter the IP Address of your router device in the address bar. This should open the login page for the Web management. Follow instructions to login and complete the configuration.













Appendix A - SIPBROKER Codes

SipBroker provides a system for dialing between SIP / VSP networks using a set of special dial codes.




The SIP Code list is referred to at: <http://www.sipbroker.com>

You can check at the SipBroker web site for details on using the system and a complete updated list.

Using the SIP codes, users can dial to the different SIP server numbers directly.

| SIP-Code | SIP Proxy | Provider name |
|--|------------------|--|
|  *010 | us.voxalot.com | VoXaLot (U.S.)  [Home] |
|  *011 | sipbroker.com | SIPBroker Alias Server [Home] |
|  *012 | public.tello.com | ISN/ITAD Routing  |
|  *013 | sipbroker.com | SIPBroker ENUM Server [Home] |
|  *018 | gtalk2voip.com | GTalk2 VoIP [Home] |
|  *027 | zero27.net | zero27.net |
|  *031 | eu.voxalot.com | VoXaLot (Europe)  [Home] |
|  *061 | au.voxalot.com | VoXaLot (Australia)  [Home] |

Legend

-  Everything is working OK.
-  Either the provider is blocking inbound calls, or there have been 5 or more unsuccessful calls in a row (in this latter case, the first successful call resets the status back to green.)
-  Insufficient call volumes to determine status.

Appendix B - Glossary

Address mask

A bit mask used to select bits from an Internet address for subnet addressing. The mask is 32 bits long and selects the network portion of the Internet address and one or more bits of the local portion. Sometimes called subnet mask.

AAL5

ATM Adaptation Layer - This layer maps higher layer user data into ATM cells, making the data suitable for transport through the ATM network.

ADSL

Asymmetric digital subscriber line.

ATM

Asynchronous Transfer Mode - A cell-based data transfer technique in which channel demand determines packet allocation.

ATM offers fast packet technology, real time; demand led switching for efficient use of network resources.

AWG

American Wire Gauge - The measurement of thickness of a wire.

Bridge

A device connects two or more physical networks and forwards packets between them. Bridges can usually be made to filter packets, that is, to forward only certain traffic. Related devices are: repeaters which simply forward electrical signals from one cable to the other, and full-fledged routers which make routing decisions based on several criteria.

Broadband

Characteristic of any network multiplexes independent network carriers onto a single cable. Broadband technology allows several networks to coexist on one single cable; traffic from one network does not interfere with traffic from another. Broadcast A packet delivery system where a copy of a given packet is given to all hosts attached to the network. Example: Ethernet.

CO

Central Office. Refers to equipment located at a Telco or service provider's office.

CPE

Customer Premises Equipment located in a user's premises

DHCP (Dynamic Host Configuration Protocol)

DHCP is software that automatically assigns IP addresses to client stations logging onto a TCP/IP network. DHCP eliminates having to manually assign permanent IP addresses to every device on your network. DHCP software typically runs in servers and is also found in network devices such as Routers.

DMT

Discrete Multi-Tone frequency signal modulation

Downstream rate

The line rate for return messages or data transfers from the network machine to the user's premises machine.

DSLAM

Digital Subscriber Line Access Multiplex

Dynamic IP Addresses

A dynamic IP address is an IP address that is automatically assigned to a client station

(computer, printer, etc.) in a TCP/IP network. Dynamic IP addresses are typically assigned by a DHCP server, which can be a computer on the network or another piece of hardware, such as the Router. A dynamic IP address may change every time your computer connects to the network.

Encapsulation

The technique used by layered protocols in which a layer adds header information to the protocol data unit (PDU) from the layer above. As an example, in Internet terminology, a packet would contain a header from the physical layer, followed by a header from the network layer (IP), followed by a header from the transport layer (TCP), followed by the application protocol data.

Ethernet

One of the most common local area network (LAN) wiring schemes, Ethernet has a transmission rate of 10 Mbps.

FTP

File Transfer Protocol. The Internet protocol (and program) used to transfer files between hosts.

Hop count

A measure of distance between two points on the Internet. It is equivalent to the number of gateways that separate the source and destination.

HTML

Hypertext Markup Language - The page-coding language for the World Wide Web.

HTML browser

A browser used to traverse the Internet, such as Netscape or Microsoft Internet Explorer.

http

Hypertext Transfer Protocol - The protocol used to carry world-wide-web (www) traffic between a www browser computer and the www server being accessed.

ICMP

Internet Control Message Protocol - The protocol used to handle errors and control messages at the IP layer.

ICMP is actually part of the IP protocol.

Internet address

An IP address is assigned in blocks of numbers to user organizations accessing the Internet. These addresses are established by the United States Department of Defense's Network Information Center. Duplicate addresses can cause major problems on the network, but the NIC trusts organizations to use individual addresses responsibly. Each address is a 32-bit address in the form of x.x.x.x where x is an eight-bit number from 0 to 255. There are three classes: A, B and C, depending on how many computers on the site are likely to be connected.

Internet Protocol (IP)

The network layer protocol for the Internet protocol suite

IP address

The 32-bit address assigned to hosts that want to participate in a TCP/IP Internet.

ISP

Internet service provider - A company allows home and corporate users to connect to the Internet.

MAC

Media Access Control Layer - A sub-layer of the Data Link Layer (Layer 2) of the ISO OSI

Model responsible for media control.

MIB

Management Information Base - A collection of objects can be accessed via a network management protocol, such as SNMP and CMIP (Common Management Information Protocol).

NAT

Network Address Translation - A proposal for IP address reuse, where the local IP address is mapped to a globally unique address.

NVT

Network Virtual Terminal

PAP

Password Authentication Protocol

PORT

The abstraction used by Internet transport protocols to distinguish among multiple simultaneous connections to a single destination host.

POTS

Plain Old Telephone Service - This is the term used to describe basic telephone service.

PPP

Point-to-Point-Protocol - The successor to SLIP, PPP provides router-to-router and host-to-network connections over both synchronous and asynchronous circuits.

PPPoE

PPP over Ethernet is a protocol for connecting remote hosts to the Internet over an always-on connection by simulating a dial-up connection.

Remote server

A network computer allows a user to log on to the network from a distant location.

RFC

Request for Comments - Refers to documents published by the Internet Engineering Task Force (IETF) proposing standard protocols and procedures for the Internet. RFCs can be found at www.ietf.org.

Route

The path that network traffic takes from its source to its destination. The route a datagram may follow can include many gateways and many physical networks. In the Internet, each datagram is routed separately.

Router

A system responsible for making decisions about which of several paths network (or Internet) traffic will follow. To do this, it uses a routing protocol to gain information about the network and algorithms to choose the best route based on several criteria known as "routing metrics".

Routing table

Information stored within a router that contains network path and status information. It is used to select the most appropriate route to forward information along.

Routing Information Protocol

Routers periodically exchange information with one another so that they can determine minimum distance paths between sources and destinations.

SNMP

Simple Network Management Protocol - The network management protocol of choice for

TCP/IP-based Internet.

SOCKET

- (1) The Berkeley UNIX mechanism for creating a virtual connection between processes.
- (2) IBM term for software interfaces that allow two UNIX application programs to talk via TCP/IP protocols.

Spanning-Tree Bridge Protocol (STP)

Spanning-Tree Bridge Protocol (STP) - Part of an IEEE standard. A mechanism for detecting and preventing loops from occurring in a multi-bridged environment. When three or more LAN's segments are connected via bridges, a loop can occur. Because a bridge forwards all packets that are not recognized as being local, some packets can circulate for long periods of time, eventually degrading system performance. This algorithm ensures only one path connects any pair of stations, selecting one bridge as the 'root' bridge, with the highest priority one as identifier, from which all paths should radiate.

Spoofing

A method of fooling network end stations into believing that keep alive signals have come from and returned to the host. Polls are received and returned locally at either end

Static IP Addresses

A static IP address is an IP address permanently assigned to computer in a TCP/IP network. Static IP addresses are usually assigned to networked devices that are consistently accessed by multiple users, such as Server PCs, or printers. If you are using your Router to share your cable or DSL Internet connection, contact your ISP to see if they have assigned your home a static IP address. User will need that address during your Router's configuration.

Subnet

For routing purposes, IP networks can be divided into logical subnets by using a subnet mask. Values below those of the mask are valid addresses on the subnet.

TCP

Transmission Control Protocol - The major transport protocol in the Internet suite of protocols provides reliable, connection-oriented full-duplex streams.

TFTP

Trivial File Transfer Protocol - A simple file transfer protocol (a simplified version of FTP) that is often used to boot diskless workstations and other network devices such as routers over a network (typically a LAN).

Telnet

The virtual terminal protocol in the Internet suite of protocols - Allows users of one host to log into a remote host and act as normal terminal users of that host.

Transparent bridging

So named because the intelligence necessary to make relaying decisions exists in the bridge itself and is thus transparent to the communicating workstations. It involves frame forwarding, learning workstation addresses and ensuring no topology loops exist (in conjunction with the Spanning-Tree algorithm).

UDP

User Datagram Protocol - A connectionless transport protocol that runs on top of TCP/IP's IP. UDP, like TCP, uses IP for delivery; however, unlike TCP, UDP provides for exchange of datagrams without acknowledgments or guaranteed delivery. Best suited for small, independent requests, such as requesting a MIB value from an SNMP agent, in which first setting up a connection would take more time than sending the data.

UNI signaling

User Network Interface signaling for ATM communications.

Virtual Connection (VC)

A link that seems and behaves like a dedicated point-to-point line or a system that delivers packets in sequence, as happens on an actual point-to-point network. In reality, the data is delivered across a network via the most appropriate route. The sending and receiving devices do not have to be aware of the options and the route is chosen only when a message is sent. There is no pre-arrangement, so each virtual connection exists only for the duration of that one transmission.

WAN

Wide area network - A data communications network that spans any distance and is usually provided by a public carrier (such as a telephone company or service provider).

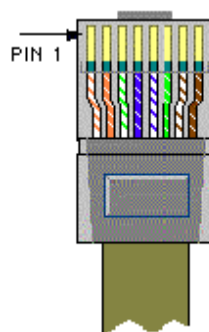
Appendix C - Cabling / Connection

Network cables connect PCs in an Ethernet network Category 5, called "Cat5" for short is commonly used type of network cable today.

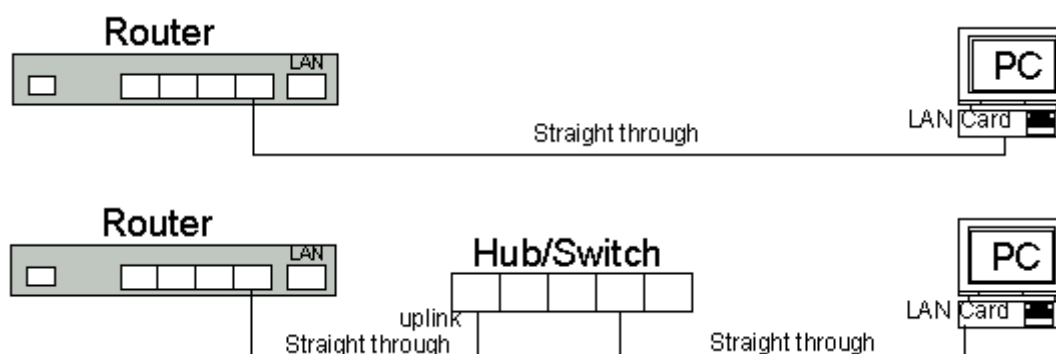
Cat 5 cables are tipped with RJ-45 connectors, which fit into RJ-45 port.

Straight-through Cables:

| Straight-through | |
|------------------|---------|
| Wire | Becomes |
| 1 | 1 |
| 2 | 2 |
| 3 | 3 |
| 6 | 6 |



LAN Connection:



Check that the network LEDs light up when you finish connecting two pieces of hardware.